IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Re: Application of: Leon et al.

Attorney Docket: 944-001.108-1

Serial No.: 10/826,687

Group Art Unit: 2457

Filed: April 16, 2004

Examiner: Uzma Alam

For: **METHOD AND DEVICE FOR PROACTIVE RATE ADAPTATION SIGNALING**

Mail Stop Appeal Brief - Patents
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

BRIEF OF APPELLANTS (37 CFR §41.37)

Sir:

This is an appeal from the Final Office Action mailed on December 16, 2009, (the "Final Office Action") rejecting claims 1-7, 9-17, 19-34 and 36.

******If any fee and/or extension is required in addition to any enclosed herewith, please charge Account No. 23-0442.***

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I. REAL PARTY IN INTEREST (37 CFR §41.37(c)(1)(i))

The real party in interest in this action is Nokia Corporation, Keilalahdentie 4, FIN-02150 Espoo, Finland, by virtue of the Assignment dated July 29, 2004 and May 11, 2004. The Assignment was recorded in the U.S. Patent and Trademark Office on August 20, 2004.

II. RELATED APPEALS AND INTERFERENCES (37 CFR §41.37(c)(1)(ii))

There are no related appeals or interferences.

III. STATUS OF CLAIMS (37 CFR§41.37(c)(1)(iii))

The status of the claims is:

Claims pending: 1-7, 9-17, 19-34 and 36.

Canceled claims: 8, 18 and 35.

Claims objected to: none.

Claims rejected: 1-7, 9-17, 19-34 and 36.

Claims on appeal: 1-7, 9-17, 19-34 and 36.

IV. STATUS OF AMENDMENTS (37 CFR §41.37(c)(1)(iv))

No amendment of claims 1-7, 9-17, 19-34 and 36 has been filed subsequent to final rejection.

V. SUMMARY OF CLAIMED SUBJECT MATTER (37 CFR §41.37(c)(1)(v))

Appellant's invention is directed to a cooperative rate adaptation model involving a server (data sender) and a client (data receiver) in a multimedia network. The server is responsible for the adaptation of the transmission rate to the reception rate and the adaptation of the sampling rate to the transmission rate. The client is responsible for setting the parameters of the server rate adaptation operating range and compensating for packet transfer delay variation (page 4, line 23 to page 5, line 2; page 12, lines 22-26; page 13, lines 10-21). The operating range for the server rate adaptation is specified by the allowable shift for any packet the server sends (page 5, lines 3-5; page 9, lines 17-22; page 10, lines 16-18). According to the present invention, the shift is defined as the time difference between the sampling time and the transmission time of the packet (page 9, lines 7-10).

The invention of independent claim 1 is directed to a method for use in a multimedia streaming network. The streaming network comprises a server configured for providing streaming data to the client (Figure 3; page 17, lines 13-17). The client has a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner (page 1, line 30 to page 2, line 16). The method comprises 1) defining in the client at least one parameter for determining a rate adaptation operating range, wherein the rate adaption operating range is used for rate adaptation between the server and the client (p.3, lines 1-8); 2) providing to the server information indicative of said at least one parameter (page 18, lines 17-23); 3) adapting in the server the data amount to a reception rate at the client based on said at least one parameter (page 18, lines 26-26-28); and 4) adjusting in the client packet transfer delay variation based on said adapting (page 4, lines 31-32; page 17, line 32 to page 18, line 4), wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server (page 18, lines 17-19; page 9, lines 7-9).

In the invention of dependent claim 2, the shift amount is substantially equal to said difference so as to allow the server to carry out said adapting based on the shift amount.

In the invention of dependent claim 3, the shift amount is greater than said difference so as to allow the server to carry out said adapting based on the shift amount.

In the invention of dependent claim 4, the parameter further comprises a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting based on the number.

In the invention of dependent claim 5, the method further comprises adapting a sampling rate to the transmission rate in the server based on said at least one parameter.

The invention of independent claim 6 is directed to a method for use in a multimedia streaming network. The streaming network comprises a server configured for providing streaming data to the client (Figure 3; page 17, lines 13-17). The client has a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner (page 1, line 30 to page 2, line 16). The method comprises 1) defining in the client at least one parameter for determining a rate adaptation operating range, wherein the rate adaption operating range is used for rate adaptation between the server and the client (p.3, lines 1-8); 2) providing to the server information indicative of said at least one parameter (page 18, lines 17-23); 3) adapting in the server the data amount to a reception rate at the client based on said at least one parameter (page 18, lines 26-26-28); and 4) adjusting in the client packet transfer delay variation based on said adapting (page 4, lines 31-32; page 17, line 32 to page 18, line 4), wherein the parameter comprises a shift amount in time indicative of a clock drift between the server and the client (page 10, lines 2-8; page 16, lines 5-10).

In the invention of dependent claim 7, the adapting comprises an adjustment of a transmission rate.

In the invention of dependent claim 9, the adapting comprises an adjustment of both a transmission rate and a sampling rate.

In the invention of dependent claim 10, the parameter further comprises a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled, and a clock shift amount indicative of a clock drift between the client and the server.

The invention of independent claims 11 is directed to a multimedia streaming network, which comprises at least a client; and a server for providing streaming data to the client, the client having a receiver buffer to compensate for a difference between data transmission amount by the server and data usage amount by the client so as to allow the

client to have sufficient amount of streaming data to play-out in a non-disruptive manner (Figure 3; page 17, lines 13-17; page 1, line 30 to page 2, line 16).

The client comprises:

a mechanism for defining at least one parameter for determining a rate adaptation operating range, and for providing information indicative of said at least one parameter to the server so as to allow the server to adapt the data amount to a reception rate at the client based on said at least one parameter (Figure 3; page 18, lines 17-23); and

a mechanism to adjust a packet transfer delay variation based on said adapting (page 17, line 32 to page 18, line 4), wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server (page 18, lines 17-19; page 9, lines 7-9) .

In the invention of dependent claim 12, the shift amount is equal to said difference so as to allow at the server to carry out said adapting based on said shift amount.

In the invention of dependent claim 13, the shift amount is greater than said difference.

In the invention of dependent claim 14, the parameter further comprises a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting.

In the invention of dependent claim 15, the server comprises an adapting mechanism for adapting a sampling rate to the transmission rate based on said at least one parameter.

In the invention of dependent claim 16, the parameter further comprises a further shift amount indicative of a clock drift between the server and the client.

In the invention of dependent claim 17, the server comprises an adapting mechanism for adjusting a transmission rate.

In the invention of dependent claim 19, the server comprises an adapting mechanism for adjusting both a transmission rate and a sampling rate.

In the invention of dependent claim 20, the server comprises a software program having at least a programming code for carrying out said adapting.

The invention of claim 21 is directed to a computer readable medium embedded with a software program (Figure 3, block 116). The software program comprises: programming code for defining in a client in a multimedia network at least one parameter for determining a rate adaptation operation range, wherein the streaming network comprises a server configured for providing streaming data to the client, the client having a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner, where information indicative to said at least one parameter is provided to the server so as to allow the server to carry out rate adaptation between the server and the client based on said at least one parameter, wherein said one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission of a packet at the server; and programming code for adjusting a packet transfer delay variation in the client for the rate adaptation (page 18, lines 17-21; page 17, line 32 to page 18, line 4; page 9, lines 7-9) .

In the invention of dependent claim 22, the shift amount is substantially equal to said difference so as to allow at the server to carry out said rate adaptation.

In the invention of dependent claim 23, the shift amount is greater than said difference so as to allow the server to carry out said rate adaptation.

In the invention of dependent claim 24, the parameter further comprises a number specifying a maximum difference between the number of bytes that have been sent and the

number of bytes that have been sampled so as to allow the server to carry out said rate adaptation.

In the invention of claim 25, the parameter further comprises a further shift amount indicative of a clock drift between the server and the client.

The invention of independent claim 26 is directed to an apparatus, which comprises:

a buffer for storing at least part of streaming data provided by a server in a multimedia streaming network to compensate for a difference between data transmission amount by the server and the data usage amount in a client so that sufficient amount of the streaming data can be played out in a non-disruptive manner (Figures 3, 60, 84; page 1, line 30 to page 2, line 16);

a mechanism for defining at least one parameter that determines a rate adaptation operating range in the server so as to allow the server to adapt the data transmission amount to a reception rate at the client based on said at least one parameter, wherein said one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server (Figure 3; page 18, lines 17-23; page 9, lines 7-9); and

a mechanism for adjusting a packet transfer delay variation based on said adapting (page 17, line 32 to page 18, line 4).

In the invention of dependent claim 27, the defining mechanism comprises a software program having at least a programming code for defining said at least one parameter.

In the invention of dependent claim 28, the adjusting mechanism comprises a software program having at least a code for adjusting the packet transfer delay variation.

In the invention of dependent claim 29, the shift amount is substantially equal to said difference so as to allow the server to carry out said adapting based on the shift amount.

In the invention of dependent claim 30, the shift amount is greater than said difference so as to allow the server to carry out said adapting based on the shift amount.

In the invention of dependent claim 31, the parameter further comprises a number specifying a maximum difference between the number of bytes that have been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting based on the number.

The invention of independent claim 32 is directed to a network element in the multimedia streaming network (Figure 3, 10). The network element comprises:

a receiving module for receiving a request from a client having a buffer for storing at least part of streaming data provided by the network element to compensate for a difference between data transmission amount by the network element and data usage amount by the client so that the client has sufficient amount of streaming data to play out in a non-disruptive manner, the request indicative of at least one parameter that determines a rate adaptation operating range in the network element, wherein said one parameter comprises a shift amount in time indicative of a clock drift between the network element and the client (Figure 3; page 17, lines 18-21; page 10, lines 2-8; page 16, lines 5-10); and

a mechanism for adapting, based on said at least one parameter, the data transmission amount from the network element to a reception rate at the client, so as to allow the client to adjust a packet transfer delay variation based on said adapting (Figure 3; page 18, lines 26-28).

In the invention of dependent claim 33, the adapting mechanism comprises a software program having at least a programming code for adapting the data transmission amount.

In the invention of dependent claim 34, the software program comprises a programming code for adjusting the transmission rate.

In the invention of dependent claim 36, the software program comprises a programming code for adjusting of both a transmission rate and a sampling rate.

VI. GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL (37 CFR §41.37(c)(1)(vi))

In the final office action, section 2 , claims 1-5, 9, 11-15, 17, 19-24 and 26-31 are rejected under 35 U.S.C. 102(e) as being anticipated by *Ravi et al.* (U.S. Patent No. 6,292,834, hereafter referred to as *Ravi*) and *Wang et al.* (U.S. Patent No. 5903,673, hereafter referred to as *Wang*) incorporated by reference.

Claim 6-7, 10, 16, 25, 32-34 and 36 are rejected under 35 U.S.C. 103(a) as being unpatentable over *Ravi*, in view of *Nilsson et al.* (U.S. Patent Application Publication No. 2005/0172028, hereafter referred to as *Nilsson*).

VII. ARGUMENT (37 CFR §41.37(c)(1)(vii))

A. The Claimed Invention

Each of the independent claims 1, 11, 21 and 26 includes the limitation that the client provides the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

Each of the independent claims 6 and 32 includes the limitation that the client provides the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a clock drift between the server and the client.

B. Cited *Ravi*, *Wang* and *Nilsson* References

At issue here is whether *Ravi* and *Wang*, incorporated by reference, disclose the claim limitation that the client provides the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

In Section D below, appellant will show that *Ravi* and *Wang*, fail to disclose the above-mentioned claim limitation.

Also at issue is whether *Ravi*, in view of *Nilsson*, discloses that the claim limitation that the client provides the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a clock drift between the server and the client.

In Section F below, appellant will show that *Ravi*, in view of *Nilsson*, fails to disclose the above-mentioned claim limitations.

C. 102 Rejection of Independent Claims 1, 11, 21 and 26

In rejecting claims 1, 11, 21, and 26, the Examiner states that *Ravi* discloses the client provides the server information indicative of at least one parameter (play time and delta play time, Figure 7A 710, 730; col.8, lines 26-35), and *Wang* discloses adjusting in the client packet transfer delay variation based on said adapting, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server. In particular, the Examiner states that *Wang* teaches transmitting a frame based on network conditions and *Ravi* teaches providing the server with information relating to the bandwidth available in the network. The server then transmits the stream at a rate based on the information that is provided by the client. According to the Examiner, *Ravi* teaches difference between a sampling time and transmission time of a packet at the server as disclosed in *Wang* – control adds to the cumulative bandwidth balance amount of the bandwidth time consumed by current frame (col.10, lines 1-67).

At section 5 of the final office action, the Examiner also states that "According to the present application, before transmittal of a stream, the data needs to be encoded by the server. The rate of encoding is called a sampling rate" (page 8, lines 4-6); "*Wang*, as incorporated by *Ravi*, teaches the server adjusts the value of Quantization (Q) to encode the stream. This value of Q is adjusted based on a bandwidth level known by the server. *Ravi* teaches this bandwidth value is provided by the client. Adjusting the Q teaches adjusting the sampling rate of the streaming data" (page 8, lines 8-12).

D. *Ravi* and *Wang* Fail to Disclose Claim Limitations in Claims 1, 11, 21 and 26

The Examiner states that *Ravi* teaches that the client provides information regarding Playtime and Delta_Playtime to the server.

According to *Ravi*, Playtime is defined as "Due time of last packet in Playout_Buffer" and Delta_Playtime is defined as "Playtime – (Playtime at last invocation of Adjust_BW_Proc)" (Figures 7A; col.8, lines 26-36). Thus, Playtime and Delta-Playtime are related to the scheduled playtime of the last packet that is currently stored in the playout buffer 366. Accordingly, Playtime and Delta-Playtime are not indicative of a shift amount which is the difference between a sampling time and a transmission time of a packet at the server. According to the present invention, the server can sample a packet at one time (sampling time) and transmit the same packet at another time (transmission time), and the difference between these two times is defined as a shift amount in time (see page 19, lines 3-12).

The Examiner further points to col.10, lines 1-67 of *Wang* to show that *Ravi*, by incorporation by reference, discloses transmitting a frame based on network condition, and adjusting the cumulative bandwidth balance amount of bandwidth time consumed by current frame. In particular, the second loop rate control 204 adds to the cumulative bandwidth and the current frame and subtract from the cumulative bandwidth balance the amount of bandwidth time consumed by the current frame. Thus, *Ravi* discloses the difference between a sampling time and transmission time of a packet at the server.

It is respectfully submitted that, in col.9, line 64 to col. 10, line 20, *Wang* discloses:

> *Loop step 406 and next step 418 define a loop in which each frame, both I-frames and P-frames, are processed according to steps 408-416. In step 408, secondary closed loop rate control 204 adjusts the cumulative bandwidth balance according to the size of the current frame. In particular, secondary closed loop rate control 204 adds to the cumulative bandwidth balance time which elapses between the previous frame and the current frame and subtracts from the cumulative bandwidth balance the amount of bandwidth time consumed by the current frame. In one embodiment, the bandwidth time is measured in terms of seconds. In particular,*

since bandwidth is expressed in an amount of data per period of time (e.g., kilobits per second), the size of the current frame, which is expressed in terms of an amount of data, divided by bandwidth results in a measure of bandwidth time consumed by the current frame. A particularly large frame, such as an I-frame for example, consumes more bandwidth time than elapses between the current frame and the preceding frame. Accordingly, secondary closed loop rate control 204 notes a reduction in the cumulative bandwidth balance. Conversely, a particularly small frame consumes less bandwidth time than elapses between the current frame and a preceding frame and results in an increase in the cumulative bandwidth balance.

In the above passage, *Wang* only describes the loop step 408, where the encoder adjusts the cumulative bandwidth balance according to the size of the current frame. In particular, a large frame such as an I-frame consumes more bandwidth time causing a reduction in the cumulative bandwidth balance whereas a small frame results in an increase in the cumulative bandwidth.

In col.10, lines 21-58, *Wang* discloses:

In test step 410, secondary closed loop rate control 204 determines whether the cumulative bandwidth balance is greater than the upper threshold of the range determined in step 404. If the cumulative bandwidth balance is within the desired range, processing transfers to test step 414 which is described more completely below. Conversely, if the cumulative bandwidth balance is greater than the desired range, excess bandwidth is accumulating and processing transfers to step 412 in which secondary closed loop rate control 204 decreases Q 114. Accordingly, video image quality is increased at the expense of increased bandwidth consumed by subsequent frames. This is appropriate since unused accumulating bandwidth is detected and using such bandwidth improves the overall perceived quality of the motion video image. In one embodiment, Q 114 is adjusted 1% for every 3% of the upper threshold that is exceeded by the cumulative bandwidth buffer. After step 412, processing of the current frame by secondary closed loop rate control 204 completes.

In test step 414, secondary closed loop rate control 204 determines whether the cumulative bandwidth balance is less than the lower threshold of the desired range determined in step 404. If the cumulative bandwidth is within the desired range, processing of the current frame by secondary closed loop rate control 204 completes. Conversely, if the cumulative bandwidth balance is below the desired range, bandwidth is being consumed at too great a rate and processing transfers to step 416 in which secondary closed loop rate control 204 increases Q 114. Accordingly, image quality is sacrificed to conserve bandwidth used by subsequent frames. Therefore, small excesses in consumed bandwidth which are undetected by primary open loop rate control 202 but which accumulate over time are detected by secondary closed loop rate control 204 and available bandwidth is not exceeded. In one embodiment, Q 114 is adjusted 1% for every 3% of the lower threshold that exceeds the cumulative bandwidth buffer. After step 416, processing of the current frame by secondary closed loop rate control 204 completes.

In the above passages, *Wang* describes how the encoder increases a quantization parameter Q if the cumulative bandwidth balance is greater than a desired range, and decreases the quantization parameter Q if the cumulative bandwidth balance is smaller than the desired range.

The above two passages have nothing to do with the client providing the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

In col.10, line 59 to col.11, line 5, *Wang* discloses:

The result of processing according to logic flow diagram 400 (FIG. 4) is a cyclical fluctuation of the cumulative bandwidth balance. Processing each I-frame, which is typically many times larger than the average P-frame, results in a sudden and dramatic decrease in the cumulative bandwidth balance to a locally minimum

value. However, each I-frame is typically followed by a number of P-frames, processing of which results in small, incremental increases in the cumulative bandwidth balance. The cumulative bandwidth balance typically has a locally maximum balance immediately prior to processing of an I-frame by secondary closed loop rate control 204 (FIG. 2). The cumulative bandwidth balance therefore fluctuates cyclically with a period which substantially coincides with the I-frame interval.

In the above passage, *Wang* describes the reason why and how the cumulative bandwidth balance fluctuates cyclically.

Again, the above passage has nothing to do with the client providing the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

At section 5 of the final office action, the Examiner states that "According to the present application, before transmittal of a stream, the data needs to be encoded by the server. The rate of encoding is called a sampling rate"; "*Wang*, as incorporated by *Ravi*, teaches the server adjusts the value of Quantization (Q) to encode the stream. This value of Q is adjusted based on a bandwidth level known by the server. *Ravi* teaches this bandwidth value is provided by the client. Adjusting the Q teaches adjusting the sampling rate of the streaming data".

Again, whether *Ravi* teaches the bandwidth value provided by the client is not relevant to the claim limitation of providing to the server information indicative of said at least one parameter, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

In summary, *Ravi* discloses that the client provides to the server information regarding Playtime and Delta_Playtime, which are only related to the scheduled playtime of the last packet currently stored in the playout buffer. *Wang* discloses that the encoder adjusts

the cumulative bandwidth balance according to the size of the current frame. Wang also describes why and how the cumulative bandwidth balance fluctuates cyclically.

The Examiner fails to explicitly point out where *Ravi* and *Wang* disclose the client providing the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

E.1 *Ravi* and *Wang* Fail to Anticipate Independent Claims 1, 11, 21 and 26

As pointed out in Subsection D above, *Ravi* and *Wang* as incorporated by reference, fail to disclose the claim limitation of providing to the server information indicative of said at least one parameter, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

For the above reasons, *Ravi* and *Wang* fail to anticipate claims 1, 11, 21 and 26.

E.2 Dependent Claims 2-5, 9, 12-15, 17, 19, 20, 22-24 and 27-31

Claims 2-5, 9, 12-15, 17, 19, 20, 22-24 and 27-31 are dependent from claims 1, 11, 21 and 26 and include further limitation. For reasons regarding claims 1, 11, 21 and 26 above, *Ravi* and *Wang* also fail to anticipate claims 2-5, 9, 12-15, 17, 19, 20, 22-24 and 27-31.

F. 103 Rejection of Independent Claims 6 and 32

In rejecting claims 6 and 32, the Examiner states that *Ravi* discloses providing to the server indicative of at least one parameter defined by the client (Playtime, Delta_Playtime, Figure 7A 710, 730; col. 8, lines 26-35). The Examiner admits that *Ravi* fails to disclose adjusting in the client packet transfer delay variation based on said adapting, wherein said at least one parameter comprises a shift amount in time indicative of a clock drift between the server and the client. The Examiner points to *Nilsson* for disclosing notifying a clock drift (paragraphs 0133-0134) and, therefore, it would have been obvious for a person skilled in the art to combine the shift of *Ravi* with the clock drift of *Nilsson* so as to prevent a buffer overrun.

In paragraph [0133], *Nilsson* discloses:

To determine the exact amount of data in the client's decoding buffer 41, the server also needs to know the timestamp of the data packet that the client is currently decoding and presenting. The server 10 calculates this using two assumptions: firstly that the client 40 starts decoding immediately after the server 10 sends the first packet; and secondly, that the client's clock does not drift significantly from the server's clock in the duration of streaming.

In paragraph [0134], *Nilsson* discloses:

In practice both assumptions have been found to be valid. The client 40 is designed to start decoding immediately on receipt of data, and so any error on the server's estimated presentation time would result in an underestimate for the amount of data in the decoding buffer 41, which as explained above is not a problem. Drift between the client's and server's clocks during a typical streaming session is most likely to be negligible compared to the amounts of data being buffered. For example, with a difference of 100 parts per million, it would take 10000 seconds, or nearly three hours, for a drift of one second to occur. In the rare case of a large amount of drift accumulating, the client 40 can warn the server 10 by use of a control message, such as the one described earlier that is sent for decoding buffer underflow.

It is respectfully submitted that, in paragraph [0133], *Nilsson* only discloses that, to determine the exact amount of data in the client's buffer, the server also needs to know the timestamp of the data packet that the client is currently decoding under the assumption that the client's clock does not drift significantly from the server' clock. In paragraph [0134], *Nilsson* discloses that, in the rare case of a large amount of drift accumulating, the client can warn the server by use of a control message, such as the one described earlier that is sent for decoding buffer underflow.

According to *Nilsson*, in case of decoding buffer underflow, the server is notified of the buffer underflow so that the server will send packets as quickly as possible (paragraph

[0126]-[0127]). Furthermore, even if the server knows what the timestamp of the data packet the client is currently decoding, the server does not know the clock drift between the server and the client. While the server may be able to estimate the cumulating data amount in the client from the timestamp of the data packet the client is currently encoding, the server may not be able to know the exact cause of data cumulating in the client. Nevertheless, *Nilsson* fails to disclose providing to the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a clock drift between the server and the client.

In summary, *Nilsson* discloses that the server is notified when a large amount of drifting accumulating and in case of client's buffer underflow so that the server can send packets as quickly as possible.

Nilsson does not disclose that the information provided to the server indicative of at least one parameter wherein the parameter comprises a shift amount in time indicative of a clock drift between the server and the client.

For the above reasons, *Ravi*, in view of *Nilsson*, fails to render claims 6 and 32 obvious.

F.2. Dependent Claims 7, 10, 16, 25, 33, 34 and 36

Dependent claims 7, 10, 16, 25, 33, 34 and 36 are dependent from claims 6 and 32 and include further limitations. For reasons regarding claims 6 and 32 above, *Ravi*, in view of *Nilsson*, also fails to render claims 7, 10, 16, 25, 33, 34 and 36 obvious.

VIII CLAIMS APPENDIX (37 CFR §41.37(c)(1)(viii))

1. A method comprising:

defining in a client in a multimedia streaming network at least one parameter for determining a rate adaptation operating range, wherein the streaming network comprises a server configured for providing streaming data to the client, the client having a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner, and wherein the rate adaption operating range is used for rate adaptation between the server and the client;

providing to the server information indicative of said at least one parameter;

adapting in the server the data amount to a reception rate at the client based on said at least one parameter; and

adjusting in the client packet transfer delay variation based on said adapting, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

2. A method according to claim 1, wherein said shift amount is substantially equal to said difference so as to allow the server to carry out said adapting based on the shift amount.

3. A method according to claim 1, wherein said shift amount is greater than said difference so as to allow the server to carry out said adapting based on the shift amount.

4. A method according to claim 1, wherein said at least one parameter further_comprises a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting based on the number.

5. A method according to claim 1, further comprising adapting a sampling rate to the transmission rate in the server based on said at least one parameter.

6. A method comprising:

defining in a client in a multimedia streaming network at least one parameter for determining a rate adaptation operating range, wherein the streaming network comprises a server configured for providing streaming data to the client, the client having a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner, and wherein the rate adaption operating range is used for rate adaptation between the server and the client;

providing to the server information indicative of said at least one parameter; adapting in the server the data amount to a reception rate at the client based on said at least one parameter; and

adjusting in the client packet transfer delay variation based on said adapting, wherein said at least one parameter comprises a shift amount in time indicative of a clock drift between the server and the client.

7. A method according to claim 6, wherein said adapting comprises an adjustment of a transmission rate.

8. (canceled)

9. A method according to claim 1, wherein said adapting comprises an adjustment of both a transmission rate and a sampling rate.

10. A method according to claim 1, wherein said at least one parameter further comprises:

a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled; and

a clock shift amount indicative of a clock drift between the client and the server.

11. A multimedia streaming network comprising:

at least a client; and

a server for providing streaming data to the client, the client having a receiver buffer to compensate for a difference between data transmission amount by the server and data usage amount by the client so as to allow the client to have sufficient amount of streaming data to play-out in a non-disruptive manner, wherein the client comprises:

a mechanism for defining at least one parameter for determining a rate adaptation operating range, and for providing information indicative of said at least one parameter to the server so as to allow the server to adapt the data amount to a reception rate at the client based on said at least one parameter; and

a mechanism to adjust a packet transfer delay variation based on said adapting, wherein said at least one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

12. A multimedia streaming network according to claim 11, wherein said shift amount is equal to said difference so as to allow at the server to carry out said adapting based on said shift amount.

13. A multimedia streaming network according to claim 11, wherein said shift amount is greater than said difference.

14. A multimedia streaming network according to claim 11, wherein said at least one parameter further comprises a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting.

15. A multimedia streaming network according to claim 11, wherein the server comprises an adapting mechanism for adapting a sampling rate to the transmission rate based on said at least one parameter.

16. A multimedia streaming network according to claim 11, wherein said at least one parameter further comprises a further shift amount indicative of a clock drift between the server and the client.

17. A multimedia streaming network according to claim 11, wherein the server comprises an adapting mechanism for adjusting a transmission rate.

18. (canceled)

19. A multimedia streaming network according to claim 11, wherein the server comprises an adapting mechanism for adjusting both a transmission rate and a sampling rate.

20. A multimedia streaming network according to claim 11, wherein the server comprises a software program having at least a programming code for carrying out said adapting.

21. A computer readable medium embedded with a software program comprising: programming code for defining in a client in a multimedia network at least one parameter for determining a rate adaptation operation range, wherein the streaming network comprises a server configured for providing streaming data to the client, the client having a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner, where information indicative to said at least one parameter is provided to the server so as to allow the server to carry out rate adaptation between the server and the client based on said at least one parameter, wherein said one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission of a packet at the server; and

programming code for adjusting a packet transfer delay variation in the client for the rate adaptation.

22. A computer readable medium according to claim 21, wherein said shift amount is substantially equal to said difference so as to allow at the server to carry out said rate adaptation.

23. A computer readable medium according to claim 21, wherein said shift amount is greater than said difference so as to allow the server to carry out said rate adaptation.

24. A computer readable medium according to claim 21, wherein said at least one parameter further comprises a number specifying a maximum difference between the number of bytes that have been sent and the number of bytes that have been sampled so as to allow the server to carry out said rate adaptation.

25. A computer readable medium according to claim 21, wherein said at least one parameter further comprises a further shift amount indicative of a clock drift between the server and the client.

26. An apparatus comprising:

a buffer for storing at least part of streaming data provided by a server in a multimedia streaming network to compensate for a difference between data transmission amount by the server and the data usage amount in a client so that sufficient amount of the streaming data can be played out in a non-disruptive manner;

a mechanism for defining at least one parameter that determines a rate adaptation operating range in the server so as to allow the server to adapt the data transmission amount to a reception rate at the client based on said at least one parameter, wherein said one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server; and

a mechanism for adjusting a packet transfer delay variation based on said adapting.

27. An apparatus according to claim 26, wherein said defining mechanism comprises a software program having at least a programming code for defining said at least one parameter.

28. An apparatus according to claim 26, wherein said adjusting mechanism comprises a software program having at least a code for adjusting the packet transfer delay variation.

29. An apparatus according to claim 26, wherein said shift amount is substantially equal to said difference so as to allow the server to carry out said adapting based on the shift amount.

30. An apparatus according to claim 26, wherein said shift amount is greater than said difference so as to allow the server to carry out said adapting based on the shift amount.

31. An apparatus according to claim 26, wherein said at least one parameter further comprises a number specifying a maximum difference between the number of bytes that have been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting based on the number.

32. A network element in the multimedia streaming network, said network element comprising:

a receiving module for receiving a request from a client have a buffer for storing at least part of streaming data provided by the network element to compensate for a difference between data transmission amount by the network element and data usage amount by the client so that the client has sufficient amount of streaming data to play out in a non-disruptive manner, the request indicative of at least one parameter that determines a rate adaptation operating range in the network element, wherein said one parameter comprises a shift amount in time indicative of a clock drift between the network element and the client; and

a mechanism for adapting, based on said at least one parameter, the data transmission amount from the network element to a reception rate at the client, so as to allow the client to adjust a packet transfer delay variation based on said adapting.

33. A network element according to claim 32, wherein said adapting mechanism comprises a software program having at least a programming code for adapting the data transmission amount.

34. A network element according to claim 33, wherein the software program comprises a programming code for adjusting the transmission rate.

35. (canceled)

36. A network element according to claim 33, wherein the software program comprises a programming code for adjusting of both a transmission rate and a sampling rate.

IX. EVIDENCE APPENDIX (37 CFR §41.37(c)(1)(ix))

There are no evidences submitted pursuant to 37 CFR §1.130, 1,131 or 1,132.

X. RELATED PROCEEDING APPENDIX (37 CFR §41.37(c)(1)(x))

There are no prior decisions rendered by a court or the Board in any proceeding identified pursuant to 37 CFR §41.37(c)(1)(ii).

CONCLUSION

Claims 1-7, 9-17, 19-34 and 36 are rejected in error. Appellant respectfully requests that the rejection of all pending claims be withdrawn.

Respectfully submitted,

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